WHAT IS CLAIMED IS:

1. A communications apparatus comprising:

an encoder for encoding a signal;

a code compression unit, coupled to the encoder, for compressing the encoded signal using a lossless scheme and a lossy scheme; and

a memory, coupled to an output of the code compression unit, for storing the compressed encoded signal.

2. The apparatus of claim 1 further comprising:

a code decompression unit, coupled to the memory, for decompressing the stored signal using a lossless scheme and a lossy scheme; and

a decoder, coupled to the code decompression unit, for decoding the decompressed signal.

3. The apparatus of claim 2 wherein the quality of the signal decompressed using the lossy scheme is improved by changing weighting factors and a tilt factor in a post filter.

4. The apparatus of claim 1 wherein the lossless scheme is used to compress parameters of the encoded signal having high inter-frame redundancy.

5. The apparatus of claim 4 wherein the parameters of the encoded signal having high inter-frame redundancy includes coefficients of a long term filter and codebook gains.

6. The apparatus of claim 1 wherein the lossy scheme is used to compress some parameters of the encoded signal having low inter-frame redundancy.

7. The apparatus of claim 6 wherein the parameters of the encoded signal having low inter-frame redundancy that are compressed include fixed codebook indices.

8. The apparatus of claim 6 wherein the parameters of the encoded signal having low inter-frame redundancy that are not compressed include adaptive codebook indices.

9. The apparatus of claim 1 further comprising a switch that enables an encoded signal received by a receiver to be compressed by the code compression unit and stored in the memory.

10. The apparatus of claim 2 further comprising a switch that enables the stored signal to be decompressed by the decompression unit and output from a transceiver.

11. The apparatus of claim 1 further comprising an operator interface unit.

12. The apparatus of claim 1 wherein the apparatus is a mobile telephone or a communication device.

13. A method for compressing a signal comprising the steps of:

converting the signal to a digital signal;

encoding the digital signal;

compressing, within a compression unit, the encoded signal using a lossless scheme and a lossy scheme; and

storing the compressed encoded signal in a memory coupled to an output of the compression unit.

14. The method of claim 13 further comprising the steps of:

decompressing, within a decompressing unit, the stored signal using a lossless scheme and a lossy scheme;

decoding, within a decoder, the decompressed signal; and

outputting the decoded signal.

15. The method of claim 14 wherein the quality of the signal decompressed using the lossy scheme is improved by changing weighting factors and a tilt factor in a post filter of the decoder.

16. The method of claim 13 wherein the lossless scheme is used to compress parameters of the encoded signal having high inter-frame redundancy.

17. The method of claim 16 wherein the parameters of the encoded signal having high inter-frame redundancy include coefficients of a long term filter and codebook gains.

18. The method of claim 13 wherein the lossy scheme is used to compress some parameters of the encoded signal having low inter-frame redundancy.

19. The method of claim 18 wherein the parameters of the encoded signal having low inter-frame redundancy that are compressed include fixed codebook indices.

20. The method of claim 18 wherein the parameters of the encoded signal having low inter-frame redundancy that are not compressed include adaptive codebook indices.

21. A method of improving quality of a lossy-compressed signal comprising the steps of:

performing a lossy compression of an uncompressed signal to yield a lossy-compressed signal;

performing a transform of the uncompressed signal from time domain to frequency domain;

decompressing the lossy-compressed signal;

performing a transform of the decompressed lossy-compressed signal from time domain to frequency domain;

comparing an absolute value of the transformed uncompressed signal to the absolute value of the transformed decompressed lossy-compressed signal;

adjusting weighting factors and a tilt factor until a minimal difference between the absolute values of the transformed signals is reached; and

applying the adjusted weighting factors and the adjusted tilt factor to the decompressed lossy-compressed signal.

22. The method of claim 21 wherein the transforms are performed using short time Fourier transforms.

23. The method of claim 21 wherein the method is performed in an AMR codec.

24. The method of claim 21 wherein the method is performed in an EFR codec.

25. The method of claim 21 further comprising the step of performing a subjective listening test to confirm the adjusted factors.

26. An apparatus for improving quality of a lossy-compressed signal comprising:

a code compression unit adapted to lossy-compress an uncompressed signal;

a code decompression unit adapted to decompress the lossy-compressed signal; and

a processor adapted to:

perform a transform of the uncompressed signal and of the decompressed lossy-compressed signal from time domain to frequency domain;

compare an absolute value of the transformed uncompressed signal to an absolute value of the transformed decompressed lossy-compressed signal; and

adjust weighting factors and a tilt factor until a minimal difference between the absolute values of the transformed signals has been reached.

27. The apparatus of claim 26 further comprising a post filter adapted to apply the adjusted weighting factors and the adjusted tilt factor to the decompressed lossy-compressed signal.

28. The apparatus of claim 27 wherein the apparatus comprises part of an EFR codec.

29. The apparatus of claim 27 wherein the apparatus comprises part of an AMR codec.

30. A method of sorting parameters of an encoded speech signal for compression comprising the steps of:

determining a degree of inter-frame redundancy of each of the parameters;

lossy compressing a first portion of the parameters, the first portion having relatively low inter-frame redundancy; and

losslessly compressing a second portion of the parameters, the second portion having relatively high inter-frame redundancy.

31. The method of claim 30 further comprising the step of not compressing a third portion of the parameters, the third portion of the parameters being selected according to pre-determined criteria irrespective of inter-frame redundancy.

32. The method of claim 30 wherein the degree of inter-frame redundancy of each of the parameters is determined by statistical analysis.

33. The method of claim 30 the second portion includes coefficients of a long term filter and codebook gains.

34. The method of claim 30 wherein the first portion includes fixed codebook indices and adaptive codebook indices.

35. A method for decompressing a signal comprising the steps of:

decompressing, within a decompressing unit, a compressed encoded digital signal using a lossless scheme and a lossy scheme;

decoding, within a decoder, the decompressed signal; and

outputting the decoded signal.

36. The method of claim 35 wherein the quality of the decompressed signal is improved by changing weighting factors and a tilt factor in a post filter of the decoder.

37. The method of claim 35 further comprising the step of losslessly inter-frame redundancy.

38. The method of claim 37 wherein the parameters of the encoded digital signal having high inter-frame redundancy include coefficients of a long term filter and codebook gains.

39. The method of claim 35 further comprising the step of lossy compressing some parameters of an encoded digital signal, the parameters having low inter-frame redundancy.

40. The method of claim 39 wherein the parameters of the encoded signal having low inter-frame redundancy include fixed codebook indices.

41. The method of claim 39 wherein the parameters of the encoded signal having low inter-frame redundancy include adaptive codebook indices.